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SPECTRAL MAXIMA SOUND PROCESSOR

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(71) Applicant(s)
THE UNIVERSITY OF MELBOURNE

(72) Inventor(s)
HUGH J. MCDERMOTT; ANDREW E. VANDALI

(74) Attorney or Agent
WATERMARK PATENT & TRADEMARK ATTORNEYS, Locked Bag 5, HAWTHORN VIC 3122

(57) Claim

1. A sound processing device comprising in combination:

means for receiving an electrical signal representing a sound signal;

filter means for providing amplitude signals corresponding to the amplitude of said sound signal in a plurality of spaced frequency channels;

means for selecting one or more of the amplitude signals according to magnitude;

and means for producing one or more output signals corresponding to said selected signals.

5. A sound processing device for producing stimulus signals for an electrode array, comprising:

means for receiving an electrical signal representing a sound signal;

filter means for providing amplitude signals corresponding to the amplitude of said sound signal in a plurality of spaced frequency channels;

means for selecting one or more of the amplitude signals according to magnitude;

memory means for storing individual stimulus current response characteristics;

means for mapping said selected amplitude signals into the stored current response characteristics and generating a corresponding current signal; and

means for communicating said corresponding current signals to an electrode array such that electrodes in a location corresponding to the frequency channel are stimulated with the corresponding current signal.

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ORIGINAL
COMPLETE SPECIFICATION
STANDARD PATENT

Application Number:

Lodged:

Invention Title: SPECTRAL MAXIMA SOUND PROCESSOR

The following statement is a full description of this invention, including the best method of performing it known to :- us

SPECTRAL MAXIMA SOUND PROCESSOR

FIELD OF INVENTION

The present invention relates to a method of processing received acoustic data, particularly but not exclusively for stimulating implanted electrode arrays.

BACKGROUND

The general technique of stimulating implanted electrode arrays is known from various disclosures, such as U.S. patent no. 4532930 to Crosby et al; U.S. patent no. 4207441 to Ricard et al; and U.S. patent no. 4611598. Such techniques generally involve implanting an electrode array into the cochlea to produce a sensation of hearing, connecting the array by direct or indirect means to a stimulation device, and modulating the stimulations in accordance with a signal. This signal is generally produced by processing in some fashion the electrical output of a microphone.

Many known processing techniques concentrate on utilising models of how the sensation of sound is detected by the brain in response to particular stimuli. Thus, the data is processed to some extent with the object of emphasising particular sorts of information in the stimulation of the electrode array.

SUMMARY OF INVENTION

It has been discovered, however, that according to the present invention an implant may be usefully stimulated based on an improved form of processing which does not rely on imposing a particular model on the processed data.

According to the present invention, electrical signals corresponding to received sound signals are processed by filter means to provide a signal corresponding to amplitude in a plurality of channels, and a selected number of said amplitude signals having the greatest amplitude are used to modulate stimuli delivered to the implanted electrode array. Preferably, the array is stimulated at a constant rate and the stimuli are delivered non-simultaneously.

BRIEF DESCRIPTION OF DRAWINGS

The invention will be described with reference to the accompanying figures, in which:

Figure 1 is a block diagram of an illustrative implanted neural stimulation system; and

Figure 2 is a block diagram of a sound processing system according to the present invention.

DETAILED DESCRIPTION

Referring to Figure 1, this illustrates in overview a system for stimulating an electrode array in accordance with a processed signal.

An electrode array 1, implanted into a cochlea, connects via cable 2 to a receiver - stimulator unit (RSU) 3. The entire implanted system may be of conventional type, such as the "Cochlear Mini-System 22".

The implanted system receives control signals and power from an external speech processor unit, preferably via a tuned coil RF system 5, 6 as illustrated. However, any alternative connection technique such as percutaneous connection may be employed.

The coil 6 carries a signal modulated by the processor 7 so as to cause the RSU 3 to stimulate the electrodes in the electrode array in the desired sequence, timing and amplitude.

The processor 7 in turn receives electrical analog signals from a microphone 8 worn by the user.

The present invention is concerned with the operation of the processor and particularly the method of processing the incoming electrical signal.

It is emphasised that while the invention is described in relation to a cochlear implant system, it is also applicable to speech processing in general, hearing aids, voice recognition, speech synthesis and tactile presentation of sound.

Referring to Figure 2, sound received by the microphone 8 produces a corresponding electrical signal. Sensitivity control 21 provides an adjustable attenuation to allow to some extent for the level of ambient sound. The signal is then pre-amplified and optionally compressed 22.

The signal is then processed by a bank 23 of parallel filters tuned to adjacent frequency channels. In a preferred embodiment there are 16 channels with centre frequencies from 250 to 5400 Hz, and the filter bank is a single chip device. Preferably, filter spacing is linear up to 1650 Hz and logarithmic beyond.

Each channel in the illustrated analog implementation includes a bandpass filter 24_n, then a rectifier 25_n and low pass filter 26_n to provide an estimate of amplitude for each channel. Preferably each low pass filter has a cut-off frequency of about 200 Hz. The output signals from each channel are then digitised.

The digitised outputs are modified by the microprocessor 27 so as to reflect the normal variation of hearing sensitivity with frequency. The set of outputs is multiplied by a set of corresponding coefficients so as to result in a slight increase in

system sensitivity at around 400 Hz, a reduction at higher frequencies, and subsequently a gradual increase to a broad peak in sensitivity at about 4 kHz.

The microprocessor then selects the six largest channel amplitudes at intervals of approximately 4ms. It is noted that this would not normally represent six different spectral peaks, as adjacent channels may share energy from a single spectral peak.

The selected amplitudes are then converted into stimulus current levels. As with known devices, the current levels corresponding to audible threshold and maximum comfortable level for each configuration of electrode stimulation in a particular patient are empirically determined and stored in a memory. The amplitudes are then mapped into the individual stimulus range for each implanted electrode set. An alternative method of converting amplitudes into stimulus levels is to vary pulse widths instead of or as well as current levels. The processor selects the appropriate active electrode for each stimulus pulse according to the frequency of the channel. The data is then encoded 32, and transmitted 33 by RF coil 6.

Microprocessor 27 is also connected to a loudness control 31, which users find convenient to use in association with the sensitivity control 21. Loudness control 31 essentially allows the current amplitude levels (and/or pulse widths) to be adjusted within a predefined range without affecting system sensitivity to the input signals.

Generally, the 16 most-apical stimulating electrode positions are allocated in tonotopic order to the 16 channels of the filterbank. The channel selection technique ensures that the maximum rate of stimulation on any electrode is 250 Hz.

It is emphasised that the six from sixteen channel system described is merely one arrangement and systems with less or more channels of filtering and with less or more channels selected are encompassed within the invention. Other rates of stimulation and alternative temporal ordering of the stimulus pulses may also provide satisfactory or improved performance.

As an alternative, the invention may be implemented using a digital signal processing (DSP) implementation. Preferably this uses a DSP56001 integrated circuit from Motorola.

One digital implementation employs a 128-point radix-2 fast fourier transform (FFT) to provide 65 discrete spectral values linearly spaced from 0 to 5.85 KHz (sampling rate = 11.7 KHz). The FFT is computed every 4ms from a 10.9 ms long time series of speech waveform samples. Each successive FFT computation therefore overlaps the previous and subsequent series by 6.9 ms.

Prior to computation of the FFT the time series is windowed by a shaping function to provide the desired spectral and temporal performance for the filter bank. The windowing function consists of a modified Daniell window (flat top with tapered sides in the frequency domain) modified by a Kaiser window (with theta = pi). It provides a 180 Hz filter bandwidth at -3 dB points. Note the FFT spectral sample spacing is 91 Hz, thus every 2nd sample is omitted leaving 32 spectral samples spaced 182 Hz apart. The DC (0 Hz) value is also omitted.

The 32 discrete spectral samples are then reduced to 16 spectral estimates by summation of power in adjacent spectral samples. The resulting 16-channel filter bank is arranged such that the lowest 8 channels have equal bandwidth and are linearly spaced. The highest 8 channels are arranged for an approximately logarithmic increase in filter bandwidth and spacing. Preferred centre frequencies of the 16 filter channels are:

274, 457, 640, 823, 1005, 1188, 1371, 1554, 1828, 2194, 2559, 2925, 3382, 3747, 4296, 5118 Hz.

Each of the 16 spectral channels is assigned to a unique stimulating electrode position in a tonotopic arrangement. Six electrodes are stimulated during each analysis period (i.e. every 4 ms). The six electrodes that are stimulated are selected based on the instantaneous amplitudes of the 16 spectral components. The six largest spectral components in each analysis period are selected, as in the analog version. The amplitudes of the six selected spectral components are transformed using a loudness growth power function and are then mapped into the dynamic range of the stimulating electrodes. The six stimuli are ordered from largest to smallest amplitudes and are presented to the implantee in quick succession every 4 ms.

It is also noted that the technique results in a relatively constant rate of stimulation, contrary to many prior techniques, however experimental evidence suggests that the perception of sound by users using the inventive speech processor is at least as good as or better than the perception of sound using other processors.

It is further noted that as no assumptions about received sound being speech are made, the system should provide improved performance over known techniques for non-speech sounds.

SELECTION OF ASSOCIATED COMPONENTS AND CIRCUIT DESIGN

In order to obtain satisfactory results with the invention, attention needs to be paid to the following points.

Microphone characteristics and frequency response will affect the quality of signal input to the processor, and some variation of the equalisation technique described above may improve performance.

Attention should also be paid to RF interference between the microphone and the transmitter coil when these are mounted on a common headset. This coupling produces components in the audio range. Signal to noise ratio may be improved by including a suitable preamplifier (e.g. gain 40dB) at the headset or microphone end of the cable.

It is also necessary to ensure that power supply is properly regulated to avoid ripple and noise.

It will be appreciated that other implementations and variations are possible within the spirit and scope of the invention.

THE CLAIMS DEFINING THE INVENTION ARE AS FOLLOWS:

1. A sound processing device comprising in combination:

means for receiving an electrical signal representing a sound signal;

filter means for providing amplitude signals corresponding to the amplitude of said sound signal in a plurality of spaced frequency channels;

means for selecting one or more of the amplitude signals according to magnitude;

and means for producing one or more output signals corresponding to said selected signals.

2. A device according to claim 1, wherein said frequency channels are linearly spaced up to about 1650 Hz and logarithmically spaced thereafter.

3. A device according to claim 2, wherein said means for selecting selects at least 4 amplitude signals.

4. A device according to claim 3, wherein said means for selecting selects six amplitude signals.

5. A sound processing device for producing stimulus signals for an electrode array, comprising:

means for receiving an electrical signal representing a sound signal;

filter means for providing amplitude signals corresponding to the amplitude of said sound signal in a plurality of spaced frequency channels;

means for selecting one or more of the amplitude signals according to magnitude;

memory means for storing individual stimulus current response characteristics;

means for mapping said selected amplitude signals into the stored current response characteristics and generating a corresponding current signal; and

means for communicating said corresponding current signals to an electrode array such that electrodes in a location corresponding to the frequency channel are stimulated with the corresponding current signal.

6. A device according to claim 5, further comprising normalising means for modifying said amplitude signals prior to said means for selecting, such that the amplitude signals are multiplied by a set of coefficients corresponding to normal variation of hearing sensitivity with frequency.

7. A device according to claim 5, wherein said means for selecting selects at least 4 amplitude signals.

8. A device according to claim 7, wherein said means for selecting selects 6 amplitude signals.

9. A device according to claim 6, wherein said current signals comprise a plurality of sets of stimuli ordered from largest to smallest amplitude.

DATED THIS 19TH DAY OF MAY, 1992

<u>THE UNIVERSITY OF MELBOURNE</u>

WATERMARK PATENT & TRADEMARK ATTORNEYS
THE ATRIUM
290 BURWOOD ROAD
HAWTHORN VICTORIA 3122
AUSTRALIA

DOC 09: AU000166.WPC

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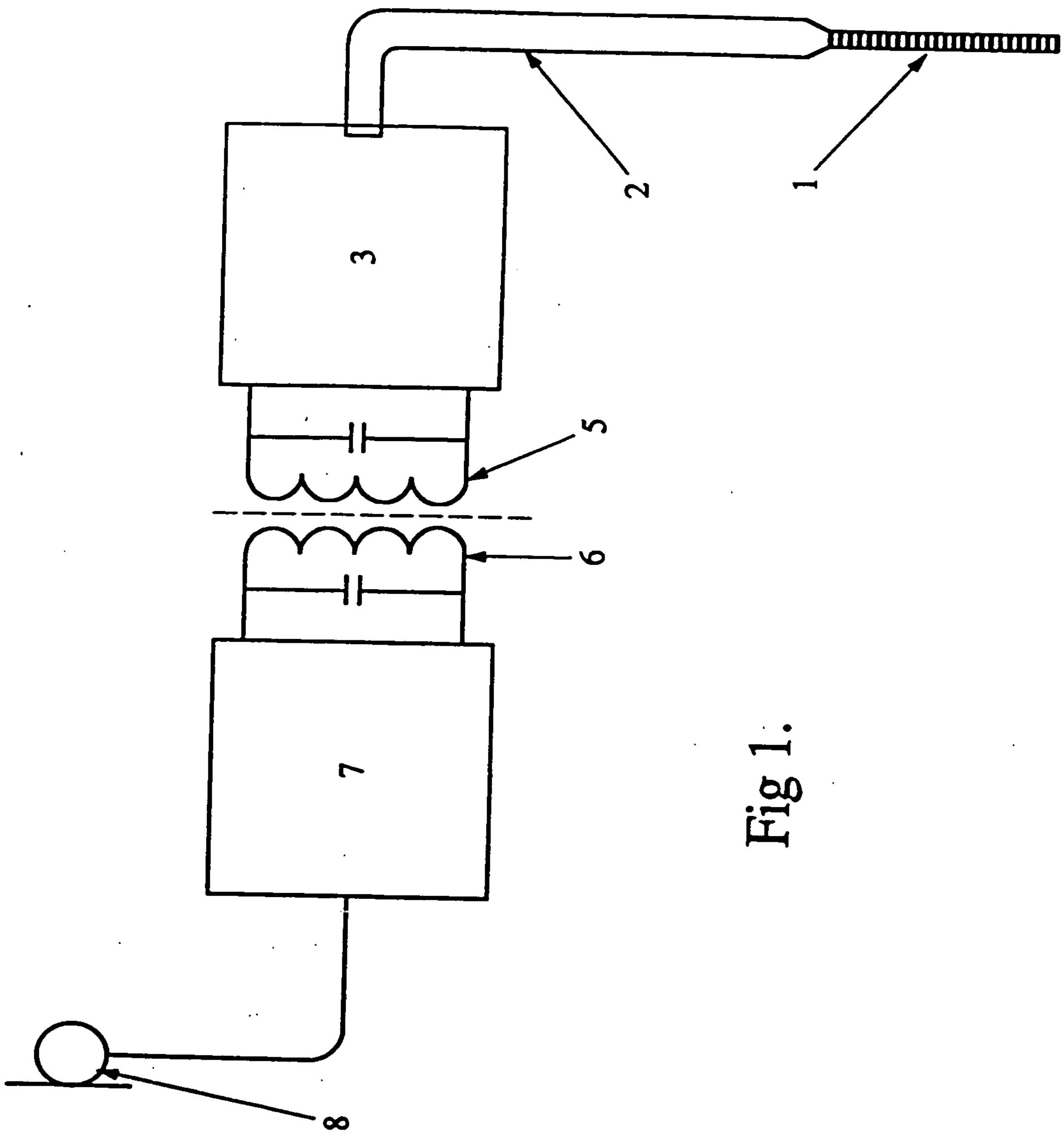

Fig 1.

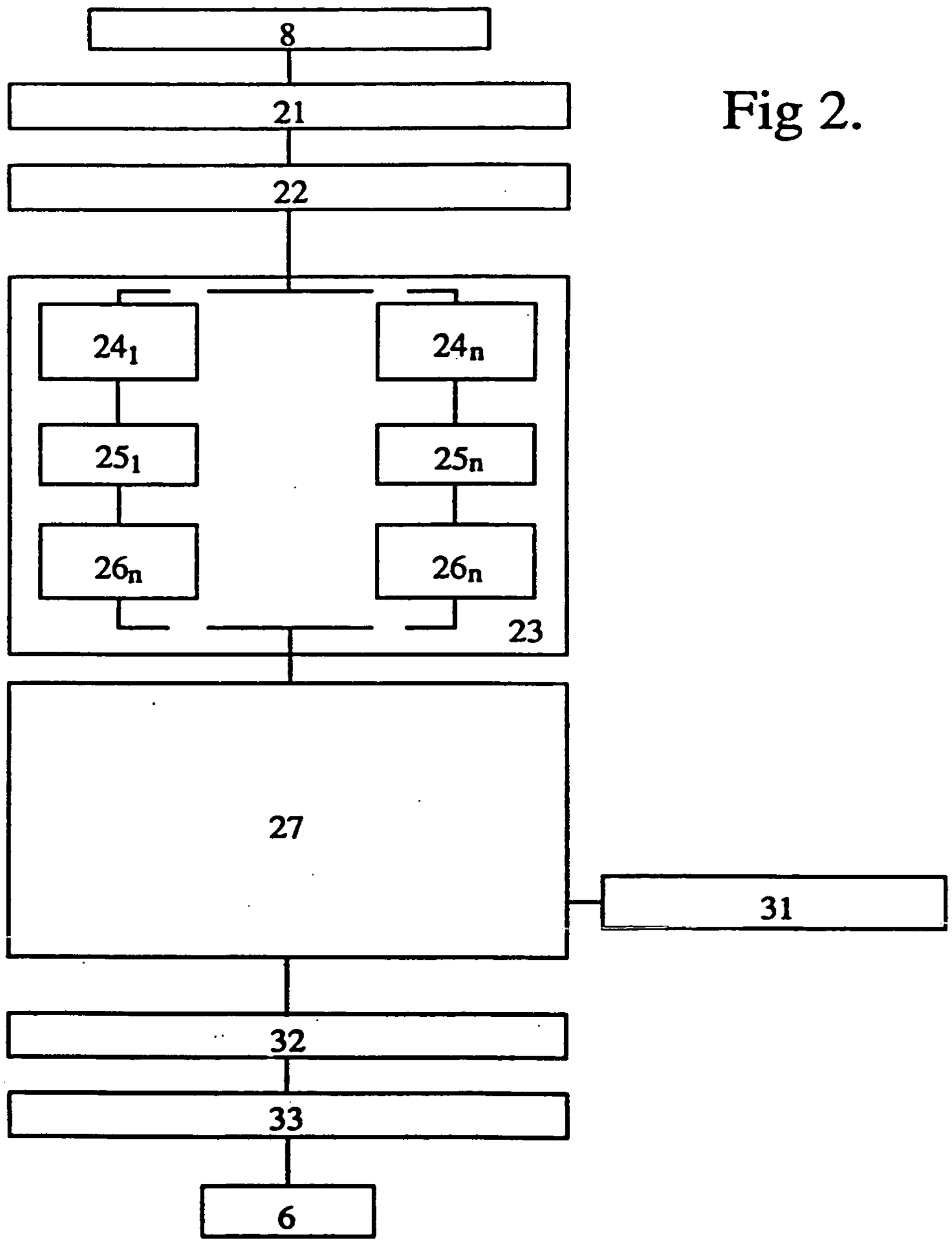

Fig 2.

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